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SIP GIN MARTINI with Open-plan Local-number Identifier Values for Enterprises (OLIVE)

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Status of this Memo

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Abstract

The Martini Working Group has defined a mechanism for SIP IP-PBX type devices to REGISTER and obtain SIP service for E.164-based Address of Records, using the GIN mechanism [draft-gin]. This document defines a means for open-plan Local-Numbers to be used with Martini-based IP-PBXs.

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1. Introduction

In many deployed SIP Service Provider (SSP) architectures, it is common to use REGISTER requests to provide the reachability information for IP-PBXs, instead of DNS-based resolution and routing. An IETF-defined mechanism for doing so is being worked on in the Martini Working Group, with the GIN mechanism [draft-gin].

The current GIN mechanism only supports E.164-based AoRs, however in actual deployments private-extension or "local" numbers are used for hosted and carrier-provided intra-Enterprise calling services. These forms of AoRs are not supported by the current GIN mechanism. This document defines a means by which they can be supported, in a manner consistent with [RFC3261] and [draft-gin].

2. Definitions

For brevity's sake, this document uses the word "request" instead of "out-of-dialog request", but in all case means out-of-dialog request.

AoR: address-of-record, as defined by RFC 3261: a URI by which the user is canonically known (e.g., on their business cards, in the From header field of their requests, in the To header field of REGISTER requests, etc.).

Open-plan: an open-plan is a dialing-plan which is not constrained to be a specific set of numbers all known to the SSP; some specific numbers may be known by the SSP, and/or a beginning set of digits are known to the SSP and used to route calls to different branches

Local-Number: an AoR which follows the form of local-number in [RFC3966], but may be encoded in a SIP or TEL URI. The local-number contains a 'phone-context' parameter identifying the scope of its number.

Implicit Registration: implicitly providing the reachability information for something other than the AoR explicitly indicated in the Register transaction.

Reachability Information: a set of URI's identifying the host and path of Proxies to reach that host; like any URI, these URI's may identify the specific connection transport, IP Address, and port information, or they may only identify FQDN's.

SSP: SIP Service Provider, as defined by [RFC5486].

3. Background

3.1. The GIN Mechanism

The GIN mechanism, defined in [draft-gin], allows a SIP UA such as an IP-PBX to Register a set of E.164 AoRs in "bulk". Instead of creating a separate REGISTER transaction for every E.164 AoR, the IP-PBX sends one REGISTER request with a 'bnc' Contact URI parameter which indicates the Contact URI needs to be expanded in the Registrar's location service database. The expansion is such that each E.164 user number becomes the user portion of the registered Contact URI, one for each implicitly registered E.164 number-based AoR.

SIP Request routing to the Registered E.164 AoR then follows normal [RFC3261] procedures, replacing the Request URI with the expanded registered Contact URI, and adding any Path information as a Route set, etc.

3.2. Local-Numbers

The Local-Number syntax for TEL URIs is defined in [RFC3966], such that a local-number is a set of digits, possibly with an extension or isdn-subaddress parameter, and scoped to the domain name or global-number-digits in its phone-context. In theory, a SIP UA can target its request to a Local-Number using the [RFC3966] syntax in a TEL URI or SIP URI, and have the request delivered to the UA or IP-PBX identified by the user digits for the given phone-context, which may subsequently route the request to the specific extension or isdn-subaddress.

In practice it’s not that simple. Most branch-office IP-PBXs do not use the Local-Number syntax for their targets, and do not recognize such syntax if they receive it in the Request URI. Often the SSP adds the phone-context to received requests from the IP-PBX, and removes it when sending to the IP-PBX. The reasons for this include: (a) the IP-PBX is wholly within the context and thus has no knowledge of, nor concern that there cold be, other contexts; (b) the IP-PBX may not actually know all the numbers in the private number plan, only its local ones; and (c) historically it has not been necessary for them to add such explicit indicators for things to work, and thus the status-quo is difficult to change.

Furthermore, Local-Numbers are difficult because they are doubly-scoped: once at the URI level by the domain name, and internally by the phone-context URI user parameter. The authoritative system for the Local-Number user portion (the system(s) which knows what they are and how to process them) is not necessarily identified by the URI’s domain name, but rather may be identified by the phone-context’s value. In other words, the SSP may not know about all possible Local-Number numbers, and even a given IP-PBX may not know them all for its Enterprise; the knowledge may be distributed. This presents difficulties for certain GIN functions such as reg-event, and is why this document refers to GIN support for Local-Numbers as being for "open-plan" scenarios.

4. The Solution - an Overview

The general concept proposed in this document is to logically apply GIN for the complete set of Local-Number "AoRs" of the Registered-to domain, as if they were individually Registered. The GIN-based REGISTER request would cause the Registrar to logically populate the set of AoR-to-Contact bindings, as it did before.

The Contact URI user portion would also be "expanded", using the same user portion as that of the implicitly registered AoRs: namely a Local-Number format. The Local-Number username is "normalized" in the same manner as [draft-gin].

Note that the list of Local-Number AoRs associated with a PBX is a matter of local provisioning at the SSP and at the PBX, as it was in [draft-gin]. The mechanism defined in this document does not provide any means to detect or recover from provisioning mismatches (although the registration event package can be used as a standardized means for auditing such AoRs).

No new option-tag is required, because this document's mechanism does not require any changes in GIN [draft-gin] registration nor in subsequent [RFC3261] routing behavior in the IP-PBX, nor in any proxies along the path. The routing follows the [RFC3261] Registered AoR-contact resolution model, which is a basic function of SIP.

The only SIP devices affected by this document's mechanism is the SSP's Registrar, which needs to update the appropriate AoR entries, and any proxy/ies of the SSP which perform route resolution by looking up the contents of the (logical) location-service database. Since such proxies may not even be in the path of the REGISTER request, an option-tag will not help. And since the Registrar and Proxies in question are all under control of the same administrative entity (the SSP), it is reasonable to expect them all to support this document's mechanism, if any do.

5. Registering for Local-Number AoRs

This document's mechanism relies on the GIN [draft-gin] Registration mechanism. The IP-PBX Registers into the SSP, using a REGISTER request with the "gin" option-tag in the Require and Proxy-Require header fields, and a Contact URI containing the "bnc" URI parameter and no user portion. After the PBX is authenticated, the registrar updates its location service so that each of the Local-Number AoRs associated with the PBX creates a unique AOR to Contact mapping.

In practice, however, the SSP domain may not have specific knowledge of any or all user names within a given phone-context's scope. In fact, the Local-Number TEL URI parameters (which are URI user parameters in SIP URIs) may only have meaning to the ultimate target of the request, or some entity which is authoritative for the phone-context's user names. Those parameters cannot be removed by the SSP if it does not actually process the user portions of the Local-Number. (i.e., if it does not have the dial-plan, etc.)

With regard to this document's mechanism, what this means is that such an SSP cannot physically instantiate an AoR in a database for every possible Local-Number and cannot physically instantiate an expanded Contact URI for every possible Local-Number user name with every possible user parameter. That does not inhibit the mechanism from working or being usable, however, because the location-service database model is purely an abstract concept. What's important is that the route-resolving Proxy be able to lookup and replace an AoR it is authoritative for, to a Registered Contact URI, such that the resultant Request URI matches what the IP-PBX expects to receive.

It is "safe" to do this because the explicitly Registered Contact URI of the [draft-gin] REGISTER request had no user portion, and thus no possible URI user parameters. As defined in [draft-gin], the Contact URI parameters of the REGISTER are saved and reused, but not URI user parameters.

There are multiple ways of describing the logical AoR instantiation and Contact URI expansion rules. They could be described as covering every possible ABNF expansion, such that every possible user and parameter logically exists in the location-service database (but obviously not physically exists). Or it could be described as only the phone-context value itself being an "AoR" entry and Contact URI expansion, with a policy to allow any and all user names and parameters to be copied instead of replaced by the Contact URI.

This remains an open issue for discussion, as discussed in section 8.

Regardless, for an implicitly Registered SIP AoR with a URI user portion matching the syntax outlined for "local-number" TEL URIs in [RFC3966]: the Contact is expanded following the other AoR models, EXCEPT that all URI user parameters are also included. For example, if the logically provisioned "AoR" from the previous examples were: "sip:12345;ext=678;phone-context=+1212555@ssp.example.com", it would logically get an automatically generated Contact value of <sip:12345;ext=678;phone-context=+1212555@198.51.100.3:5060;foo=bar> and if the AoR were "sip:12345;ext=678;phone-context=ssp.example.com@ssp.example.com", the resultant Contact value would be <sip:12345;ext=678;phone-context=ssp.example.com@198.51.100.3:5060;foo=bar>.

Note that in practice it is not uncommon to receive a SIP URI which does not strictly comply with the formatting rules of [RFC3966], but is processed as if it were, based on local policies. That is legal, of course, but from a logical perspective the SIP URI is actually retargeted or transformed into the syntactically valid form following [RFC3966], and that form MUST be the one used for routing, Contact URI expansions, etc. Likewise, if the URI were a TEL URI, it MUST be logically transformed into a SIP URI of the SSP's domain as defined in section 19.1.6 of [RFC3261], with an appropriate phone-context, before executing the rules.

As in [draft-gin], aside from the "bnc" parameter, all URI parameters present in the Contact URI in the REGISTER message MUST be copied to the Contact value stored in the location service.

Note that the location service database, and any entry model described here, is purely an abstract concept used by [RFC3261], [draft-gin], and this document; an actual implementation may do whatever it likes internally, so long as the external behavior follows the model. For example, if an SSP does not maintain any specific knowledge of the Local Number dial-plan, but simply performs prefix or default routing for an Enterprise's private extensions, the SSP could just route based on the E.164 phone-context field value without having a separate physical "AoR" database entry for each local number of that context.

6. SSP Processing of Inbound Non-E.164 Requests

The SSP Proxy/Registrar (or equivalent entity) performs traditional Proxy/Registrar behavior, based on the logical mapping described in Section 5 and [draft-gin].

7. Interaction with Other Mechanisms

The following sections describe the means by which this mechanism interacts with relevant REGISTER-related extensions currently defined by the IETF.

Currently, the descriptions are somewhat informal, and omit some details for the sake of brevity. If the MARTINI working group expresses interest in furthering the mechanism described by this document, they will be fleshed out with more detail and formality.

7.1. Globally Routable User-Agent URIs (GRUU)

The GRUU mechanism for this document's mechanism works exactly the same way as defined in [draft-gin]. The GIN GRUU mechanism has no dependency on the AoR being an E.164.

7.2. Registration Event Package

The Registration Event Packet behavior for this document's mechanism works exactly the same way as defined in [draft-gin]. The [draft-gin] reg-event model has no dependency on the AoR being an E.164.

There is, however, an issue for Local-Numbers, if the SSP does not actually know the full list of Local-Number user names in the given phone-context scope. In such a case, it is TBD for how to handle this.

This remains an open issue for discussion, as discussed in section 8.

7.3. Non-Adjacent Contact Registration (Path) and Service Route Discovery

The Path and Service-Route behavior and considerations for this document's mechanism are exactly the same as defined in [draft-gin]. The [draft-gin] Path and Service-Route model has no dependency on the AoR being an E.164.

8. Open Issues

This document has several open issues, which were noted previously. They center around the handling of Local-Numbers. Local-Numbers are difficult because they are doubly-scoped: once at the URI level by the domain name, and internally by the phone-context URI user parameter. The authoritative system for the Local-Number user portion (the system(s) which knows what they are and how to process them) is not necessarily identified by the URI's domain name, but rather may be identified by the phone-context's value.

If the phone-context identifies the SSP domain, all's well - but that's rarely the case. More likely is that it identifies an E.164 number, or a sub-domain of the SSP, or another domain entirely. This causes issues with certain functions such as the reg-event package, which has been identified as an open issue.

9. Examples

These will be fleshed out more in later versions of the draft, with explanations of the processing performed at each step. For the time being, they just show the basic syntax described above.

9.1. Usage Scenario: Basic Registration case

This example shows a basic bulk REGISTER transaction, followed by an INVITE addressed to one of the registered terminals, for a Local-Number AoR.

```
Internet                        SSP                               PBX
|                                |                                  |
|                                |REGISTER                          |
|                                |Contact:<sip:198.51.10.3;f=b;bnc>|
|                                |<---------------------------------|
|                                |                                  |
|                                |200 OK                            |
|                                |--------------------------------->|
|                                |                                  |
|INVITE                          |                                  |
|sip:1234;ext=678                |                                  |
| ;phone-context=+1212555        |                                  |
| @ssp.example.com               |                                  |
|------------------------------->|                                  |
|                                |                                  |
|                                |INVITE                            |
|                                |sip:1234;ext=678                  |
|                                | ;phone-context=+1212555          |
|                                | @198.51.100.3;f=b                |
|                                |--------------------------------->|
```

```
REGISTER sip:ssp.example.com SIP/2.0
Via: SIP/2.0/UDP 198.51.100.3:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
To: <sip:pbx123@ssp.example.com>
From: <sip:pbx123@ssp.example.com>;tag=a23589
Call-ID: 843817637684230998sdasdh09
CSeq: 1826 REGISTER
Proxy-Require: gin
Require: gin
Supported: path
Contact: <sip:198.51.10.3:5060;f=b;bnc>
Expires: 7200
Content-Length: 0
```

```
INVITE sip:1234;ext=678;phone-context=+1212555
    @ssp.example.com SIP/2.0
Via: SIP/2.0/UDP foo.example;branch=z9hG4bKa0bc7a0131f0ad
Max-Forwards: 69
To: <sip:2145550105@some-other-place.example.net>
From: <sip:alice@rabbithole.example.org>;tag=456248
Call-ID: f7aecbfc374d557baf72d6352e1fbcd4
CSeq: 24762 INVITE
Contact: <sip:line-1@192.0.2.178:2081>
Content-Type: application/sdp
Content-Length: ...

<sdp body here>
```

```
INVITE sip:1234;ext=678;phone-context=+1212555
    @198.51.100.3;f=b SIP/2.0
Via: SIP/2.0/UDP foo.example;branch=z9hG4bKa0bc7a0131f0ad
Via: SIP/2.0/UDP ssp.example.com;branch=z9hG4bKa45cd5c52a6dd50
Max-Forwards: 68
To: <sip:2145550105@some-other-place.example.net>
From: <sip:alice@rabbithole.example.org>;tag=456248
Call-ID: 7ca24b9679ffe9aff87036a105e30d9b
CSeq: 24762 INVITE
Contact: <sip:line-1@192.0.2.178:2081>
Content-Type: application/sdp
Content-Length: ...

<sdp body here>
```

10. IANA Considerations

This document makes no request of IANA.

11. Security Considerations

This section is still TBD, but it should follow/have the same issues as [draft-gin].

12. Acknowledgements

Thanks to Adam Roach for providing text copied from [draft-gin].

13. Informative References

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Appendix A - Why Local-Numbers may need processing by SSPs

There is some debate about how a non-E.164 AoR could even be received by the SSP for processing to begin with. This section describes how such could be the case.

It should be noted that this document only deals with SIP AoRs of the same URI domain name as that of the REGISTER's To URI - namely the SSP's domain.

A SIP Request targeted to a Local-Number could require processing by the SSP because:

- The SSP provides IP-Centrex type services for some of the AoRs of an Enterprise, for example for small branches, while providing SIP-Trunk service to the main IP-PBX(s). Requests from the IP-Centrex UAs will thus be targeted to Local-numbers as they are received by the SSP Proxy on their way to the IP-PBX.
- The SSP provides inbound extension dialing, for example by offering private calling-card services, such that a E.164 number call is terminated by an Application Server of the SSP which authenticates the caller belongs to an Enterprise and then allows private extension dialing, as a UAC, thereby originating a new SIP session Request using a Local-Number target.
- The SSP provides inter-branch private dialing, by routing on some number of leading digits of a Local-Number.

There are other possibilities as well, of course, but this section is only intended to provide some basic rational for why it is possible for a local-number AoR to be used and appear in the SSP.